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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)
	10/695,125	SINGHAL, MANOJ
	Examiner	Art Unit
	Douglas C. Godbold	2626

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 26 June 2007.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-6 and 8-24 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-6 and 8-24 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 28 October 2003 is/are: a)☐ accepted or b)☒ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

1. This office action is in response to correspondence filed June 26, 2007 in reference to application 10/695,125. Claims 1-6, and 8-24 are pending in the application and have been examined. Claim 7 has been cancelled.

Response to Amendment

2. The amendments to claims 1-24 have been considered and accepted. Claim 7 has been cancelled without prejudice. The rejections of claims 1, 3, 4, 10, 11, 13, 14, and 16 under 35 U.S.C 102(b) have been removed in light of these amendments. It is also noted that the 35 U.S.C. 112 rejections of claims 5 and 6 were not addressed and nor were the objections to the drawings.

Response to Arguments

3. Applicant's arguments filed June 26, 2007 have been fully considered but they are not persuasive. The applicant submits that Benyassine does not even teach a packet or packetization, much less the claimed “turning on a flag in a header of a packet of digital audio information". The examiner respectfully disagrees. Benyassine teaches packet or packetization (The encoder 112 segments the digitized speech signal into frames to generate a bitstream. In one embodiment, the speech coding system 100 uses frames having 160 samples and corresponding to 20 milliseconds per frame at a sampling rate of about 8000 Hz. The encoder 112 provides the frames via a bitstream to the communication medium 104; column 3, line 56. Further, Benyassine teaches

communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data. A window is then the basis for a packet in this use in his kind of communication medium.), and "turning on a flag in a header of a packet of digital audio information" (all flags used to mark audio frames are shown in Table 1, column 9).

Drawings

4. The drawings are objected to because they are hard to read and understand. Cleaner drawings should be submitted. Corrected drawing sheets in compliance with 37 CFR 1.121(d) are required in reply to the Office action to avoid abandonment of the application. Any amended replacement drawing sheet should include all of the figures appearing on the immediate prior version of the sheet, even if only one figure is being amended. The figure or figure number of an amended drawing should not be labeled as "amended." If a drawing figure is to be canceled, the appropriate figure must be removed from the replacement sheet, and where necessary, the remaining figures must be renumbered and appropriate changes made to the brief description of the several views of the drawings for consistency. Additional replacement sheets may be necessary to show the renumbering of the remaining figures. Each drawing sheet submitted after the filing date of an application must be labeled in the top margin as either "Replacement Sheet" or "New Sheet" pursuant to 37 CFR 1.121(d). If the changes are not accepted by the examiner, the applicant will be notified and informed of any required

corrective action in the next Office action. The objection to the drawings will not be held in abeyance.

Claim Rejections - 35 USC § 112

5. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

6. Claims 5 and 6 are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

7. Claims 5 and 6 recites the limitation "said transmitting", in claim 1. There is insufficient antecedent basis for this limitation in the claim. Claim 1 does not recite said transmitting. However for the purposes of examination, claims 5 and 6 will be assumed to be dependent of claim 8. Claims 5 and 6 should also be renumbered to fall after claim 8.

Claim Rejections - 35 USC § 103

8. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

9. The factual inquiries set forth in *Graham* **v.** *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

10. Claims 1, 3, 4, 7, 9-11, 13-16, and 20-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders (Real-Time Discrimination of Broadcast Speech/Music) in view of Benyassine (US Patent 6,694,293).

11. Consider claim 1, Saunders teaches a method for classifying an audio signal (we describe a technique which is successful at discriminating speech from music; page 993, column 1, line 1), the method comprising:

receiving an audio signal to be classified (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2);

analyzing selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43);

recording a result of analysis of the selected audio signal components (would be inherent in order to compare it);

comparing the recorded result of analysis to a threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43); and

classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43).

Saunders not specifically teach wherein classifying the audio signal further comprises turning on a flag in a header of a packet of digital audio information, wherein the flag provides an indication of classification of the audio signal based upon comparison of the recorded result of analysis and the threshold value.

In the same field of music and speech discrimination Benyassine teaches turning on a flag in a header of a packet (The encoder 112 segments the digitized speech signal into frames to generate a bitstream. In one embodiment, the speech coding system 100 uses frames having 160 samples and corresponding to 20 milliseconds per frame at a sampling rate of about 8000 Hz. The encoder 112 provides the frames via a bitstream to the communication medium 104; column 3, line 56. A window is the same as a packet in this use in the communication medium. Communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data.) of digital audio information (all flags used to mark audio frames are shown in Table 1, column 9), wherein the flag provides an indication of classification of the audio signal based upon comparison of the

recorded result of analysis and the threshold value (The music detection flag F_M is set if either threshold for music conditions are met; column 7 line 37).

Therefore it would have been obvious to one of ordinary skill in the art at the time of invention to include a detection flag as taught by Benyassine with the speech and music discrimination method of Saunders in order to provide a method to pass the classification from one device to another.

12. Consider claim 3, Saunders teaches the method according to claim 1, wherein analyzing the selected audio signal components comprises counting zero point transitions of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings).

13. Consider claim 4, Saunders teaches the method according to claim 1, wherein recording a result of analysis of the selected audio signal components comprises recording a count value of a number of zero point transitions of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings. This number would inherently have to be stored somewhere in order to process it or manipulate it).

14. Consider claim 9, Benyassine teaches the method according to claim 1, wherein classifying the audio signal occurs at a transmitting end of an audio transmission system (Figure 1, encoder 112, part of transmission side, may contain a music classifier with voice activity detector; column 3, line 62.).

15. Consider claim 10, Saunders teaches the method according to claim 1, wherein classifying the audio signal occurs at a receiving end of an audio transmission system (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2).

16. Consider claim 11, Saunders teaches the method according to claim 1, wherein the audio signal is one of an analog signal and a digital signal (A sample rate of 16Khz was chosen for this discrimination technique; page 995, column 1 line 1. If something is sampled it is well understood that it is being converted to a digital signal. this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. This further tells us that the signal started out as an analog signal as at the time of the publication of Saunders all FM broadcasts were analog.).

17. Consider claim 13, Saunders teaches the method according to claim 1, wherein the threshold value used in the comparison determined through trial and error of a plurality of iterations in a comparing device (Data was collected manually by listening, collecting and storing features, and labeling the segment. A variety of content was

processed, including talk, commercials, and many types of music. Once the classifier was trained, the parameters were stored and fed into the real-time feature extraction/classifier routine; page 995, column 1, line 33).

18. Consider claim 14, Saunders teaches the method according to claim 1, wherein analyzing selected audio signal components comprises counting zero point transitions of the audio signal for a predetermined period of time (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings).

19. Consider claim 15, Benyassine teaches the method according to claim 1, further comprising:

converting the audio signal from an analog signal to a digital signal (figure 1, A/D converter 108);

encoding the audio signal (figure 1, encoder 112);

packetizing the audio signal (communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data);

transmitting the audio signal (figure 1, signals are transmitted over communication medium 104);

decoding the audio signal (using decoder 114, figure 1); and

processing the audio signal, wherein processing at least comprises one of storing the audio signal and playing the audio signal (output of system is synthesized speech signal 120, figure 1).

20. Consider claim 16, Saunders teaches an apparatus for classifying an audio signal (The experimental setup used a Gradient A/D unit attached to a workstation; page 995, column 1, line 38), the apparatus comprising:

a zero point counter for counting and recording zero point transitions encountered in analysis of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43); and

a comparator for comparing a recorded result of analysis to a threshold value and classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43).

However Saunders does not specifically teach a circuit for packetizing the audio signal into packets, said packets including a header, said header including a flag indicating classification of the audio signal.

In the same field of music and speech discrimination Benyassine teaches a circuit for packetizing the audio signal into packets (The encoder 112 segments the digitized speech signal into frames to generate a bitstream. In one embodiment, the

speech coding system 100 uses frames having 160 samples and corresponding to 20 milliseconds per frame at a sampling rate of about 8000 Hz. The encoder 112 provides the frames via a bitstream to the communication medium 104; column 3, line 56. A window is the same as a packet in this use in the communication medium. communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data), said packets including a header, said header including a flag indicating classification of the audio signal (all flags used to mark audio frames are shown in Table 1, column 9. The music detection flag F_M is set if either threshold for music conditions are met; column 7 line 37).

21. Consider claim 20, Benyassine teaches the apparatus according to claim 16, further comprising at least one of an audio signal encoder (figure 1, encoder 112) and an audio signal decoder (figure 1, decoder 114).

22. Consider claim 21, Benyassine teaches the apparatus according to claim 20, further comprising a speech/music classifying device being associated with the audio signal encoder (Figure 1, encoder 112, part of transmission side, may contain a music classifier with voice activity detector; column 3, line 62.).

23. Consider claim 22, Saunders teaches the apparatus according to claim 20, further comprising a speech/music classifying device being associated with the audio

signal decoder (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. An FM signal must be decoded before it can be classified or played or manipulated in anyway).

24. Consider claim 23, Saunders teaches the apparatus according to claim 20, further comprising a signal processor and an audio processing unit associated with the audio signal decoder (The experimental setup used a Gradient A/D unit attached to a workstation; page 995, column 1, line 38. Using data processed on the fly and tuning the radio dial at will, the classification accuracy averaged between 95 and 96%; page 995, column 1, line 43. This is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. An FM signal must be decoded before it can be classified or played or manipulated in anyway).

25. Consider claim 24, Benyassine teaches the apparatus according to claim 20, further comprising a bitstream multiplexer associated with the audio signal decoder (signal (communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data. It is inherent that some kind of multiplexing must be employed in order to packetize the data).

26. Claims 2 and 17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Benyassine as applied to claims 1 and 16 above and further in view of Carey (A Comparison of Features for Speech, Music Discrimination).

27. Consider claim 2, Saunders in view of Benyassine teaches the method according to claim 1, but does not specifically teach wherein classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value further comprises:

if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music; and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech.

In the same field of speech/music discrimination, Carey teaches if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music (table 1 shows that the mean value of number of zero crossing (u) for music 0.18 is greater than that of speech 0.17); and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech (table 1 shows the mean value of zero crossing for speech 0.17 was less that music 0.18).

Although Saunders in view of Benyassine uses a slightly different zero crossing analysis method than does Carey, it would have been obvious to one of ordinary skill in

the art at the time of the invention to use the parameters of Carey as this method would be computationally inexpensive (Carey page 151, column 2, section 4.4).

28. Consider claim 17, Saunders in view of Benyassine teaches the apparatus according to claim 16, but does not specifically teach wherein classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value in the comparator further comprises:

if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music; and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech.

In the same field of speech/music discrimination, Carey teaches if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music (table 1 shows that the mean value of number of zero crossing (u) for music 0.18 is greater than that of speech 0.17); and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech (table 1 shows the mean value of zero crossing for speech 0.17 was less that music 0.18).

Although Saunders in view of Benyassine uses a slightly different zero crossing analysis method than does Carey, it would have been obvious to one of ordinary skill in the art at the time of the invention to use the parameters of Carey as this method would be computationally inexpensive (Carey page 151, column 2, section 4.4).

29. Claims 5, 6, 8, 18 and 19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Benyassine as applied to claims 20 and above, and further in view of Pohlmann (Principles of Digital Audio).

30. Consider claim 8, Saunders teaches the method according to claim 1, further comprising:

selecting a number of transmitted audio signal components for analysis (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43.).

However Saunders does not specifically teach transmitting components of the audio signal having a frequency less than a predetermined frequency.

In the same field of audio analysis, Benyassine teaches transmitting an audio signal using encoder 112 of figure 1 that samples at a rate of 8000Hz; column 3, line 60.

Therefore it would have been obvious to combine the sampling of audio for transmitting of Benyassine with the classification system of Saunders in order to allow the transmissions of digital signals.

This does not say specifically that that the audio being transmitted is less than a predetermined frequency.

In the same field of audio encoding, Pohlmann teaches that sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section.

Therefore it would have been obvious to combine the sampling of Benyassine with the filtering of Pohlmann in order to prevent aliasing, and to provide a way to digitize the audio signal for analysis, coding and transmission.

31. Consider claim 5, Pohlmann teaches, the method according to claim 8, wherein transmitting components of the audio signal having a frequency less than a predetermined frequency comprises passing the audio signal through a low pass filter, the low pass filter being adapted to permit transmission of frequencies below the predetermined frequency (sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section. This is made necessary by the sampling of Benyassine column 3, line 60.).

32. Consider claim 6, Pohlmann teaches, the method according to claim 1, wherein selecting a number of transmitted audio signal components for analysis comprises passing transmitting digital audio components through a decimator, wherein every 1 in N audio signal components is transmitted and audio signal components between 1 and N are discarded. (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ... For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

33. Consider claim 18, Saunders teaches the apparatus according to claim 16, but does not specifically teach further comprising:

a low pass filter for preventing transmission of components of the audio signal having a frequency greater than a predetermined frequency; and

a decimator for selecting a reduced number of audio components for analysis.

In the same field of audio analysis, Benyassine teaches transmitting an audio signal using encoder 112 of figure 1 that samples at a rate of 8000Hz; column 3, line 60.

Therefore it would have been obvious to combine the sampling of audio for transmitting of Benyassine with the classification system of Saunders in order to allow the transmissions of digital signals.

This does not say specifically that that the audio being transmitted is less than a predetermined frequency nor the use of a decimator.

In the same field of audio encoding, Pohlmann teaches that sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section.

Therefore it would have been obvious to combine the sampling of Benyassine with the filtering of Pohlmann in order to prevent aliasing, and to provide a way to digitize the audio signal for analysis, coding and transmission.

This combination does not teach specifically a decimator. But later in the book, Pohlmann teaches a decimator for selecting a reduced number of audio components for analysis. This is nothing more than resampling the audio signal As noted many different

sampling rates are used, devices cannot be connected when their sampling rates differ... For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph.

Therefore it would have been obvious to one of ordinary skill in the art to include the decimating as taught by Pohlmann with the system of Saunders and Benyassine in order to provide a method for being able to connect different devices with different sampling rates (Pohlmann page 460, first full paragraph).

34. Consider claim 19, Pohlmann teaches the apparatus according to claim 18, wherein the decimator selecting a reduced number of audio components for analysis comprises the decimator selecting every 1 in N audio signal components to be transmitted and selecting the audio signal components between 1 and N to be discarded (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ... For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph. This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

35. Claim 12 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders. Saunders teaches the method according to claim 1, but does not specifically

teach wherein the threshold value used in the comparison is pre-determined and pre-set by a user.

However Saunders does teach Data was collected manually by listening, collecting and storing features, and labeling the segment. A variety of content was processed, including talk, commercials, and many types of music. Once the classifier was trained, the parameters were stored and fed into the real-time feature extraction/classifier routine; page 995, column 1, line 33.

With data being collected manually, it must be entered manually, and although is not specifically the threshold, one of ordinary skill in the art that the training of the classifier by manually collecting data is changing the threshold. Therefore in fact, the user is in a way changing the threshold value is preset and determined by the user.

Conclusion

36. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of

the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Douglas C. Godbold whose telephone number is (571) 270-1451. The examiner can normally be reached on Monday-Thursday 7:00am-4:30pm Friday 7:00am-3:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached on (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DCG

Patrick N. Edouard
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